CLAIMS

1/ A method of improving sound playback of digitized speech signals transmitted to a telecommunications terminal at the beginning of a telephone call set up over a communications network where the signals are transmitted in the form of packets, and in particular at the beginning of a VOIP call set up under Internet protocol, in the event said call is set up from a sending telecommunications terminal which is fitted with voice activity detection means so as to be capable of transmitting only those digitized sound signal packets that actually contain speech, which packets are taken from a set of sound signal packets that are available for being transmitted after the sound has been digitized and encoded in the sending terminal, the method providing for sound signal packets to be transmitted from the digitizing and encoding means without taking account of the presence or absence of speech signals in the processed sound signals during an initial stage of call optimization.

2/ A method according to claim 1, in which the initial telephone call optimization stage during which digitized sound signal packets are transmitted from a sending terminal without taking account of the presence or absence of speech signals in the processed signals is of a duration that is selected in such a manner as to enable a receiving terminal to receive a sufficient number of digitized sound signal packets relating to the call to enable the size of the receive buffer for digitized sound signal packets to be determined on the basis of a statistical evaluation of the delays observed on the received packets.

3/ Telecommunications hardware, in particular a subscriber terminal or a common terminal, which is connected to a network enabling packets to be exchanged

and which is designed to communicate over the network with a compatible terminal by means of digitized sound signal packets including digitized speech signals produced in the context of a VOIP type telephone call that is set up over the network under IP protocol or an equivalent protocol, the hardware comprising means in a programmed control unit enabling a number of digitized sound signal packets to be transmitted when a telephone call is set up and during an initial optimization stage, said number being sufficient to enable a receiver terminal to determine the size of a receive buffer for digitized sound signal packets by statistically evaluating the delays observed on the received packets, and voice activity determining means enabling digitized sound signals to be transmitted only if they contain speech signals, said voice activity determining means being prevented from acting until the initial optimization stage has terminated.

4/ Telecommunications hardware, in particular a subscriber terminal or a common terminal, according to claim 3, having timing means in a programmed control unit acting on the voice activity determining means of the terminal so that said means act only after the end of an initial optimization stage of determined duration.

5/ Telecommunications hardware, in particular a subscriber terminal or a common terminal, according to claim 4, in which the timing means act to temporarily inhibit the action of the voice activity determining means until the end of the initial stage of call optimization.